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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/651,520	08/29/2003	David P. Helm	CM05114H	8879

20576 7590 02/22/2008

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EXAMINER	
HASHEM, LISA	
ART UNIT	**PAPER NUMBER**
2614	
MAIL DATE	**DELIVERY MODE**
02/22/2008	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)
	10/651,520	HELM ET AL.
	Examiner	Art Unit
	Lisa Hashem	2614

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *16 November 2007*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1,2,4-7,10,12-14,16,17 and 22-30* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1,2,4-7,10, 12-14,16,17 and 22-30* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

FINAL DETAILED ACTION

Response to Arguments

1. Applicant's arguments with respect to claims 1, 2, 4-7, 10, 12-14, 16, 17, and 22-30 have been considered but are moot in view of the new ground(s) of rejection.

Claim Rejections - 35 USC § 102

2. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

3. Claims 1, 4-7, 14, 16, 17, 22, 23, and 25-30 are rejected under 35 U.S.C. 102(e) as being anticipated by U.S. Pat. No. 6,360,271 by Schuster et al, hereinafter Schuster.

Regarding claim 1, Schuster discloses a packet switched communications system (Fig. 2; col. 1, lines 22-28; col. 2, lines 10-14) having a dynamic voice jitter buffer (Fig. 2, 34; i.e. jitter buffer; col. 3, lines 59-67) for use with voice over Internet protocol (VoIP) packets (col. 2, lines 10-14; col. 8, lines 43-56) comprising: a source (Fig. 2, 12; i.e. transmitter) transmitting a plurality of VoIP packets (i.e. real-time media; packet stream) for a call (col. 8, lines 44-48; col. 9, lines 7-49);

a destination (Fig. 2, 20; i.e. receiver) for receiving the plurality of VoIP packets for the call (col. 8, lines 48-52; col. 9, line 66 – col. 10, line 6); and

at least one router (i.e. processing device; router) for routing the plurality of VoIP packets for the call from the source to the destination (col. 14, lines 33-51),

wherein at least one of the plurality VoIP packets for the call conveys congestion information (i.e. varying transmission delays; jitter) regarding the packet switched communications system (col. 10, lines 2-11), and wherein a size of the dynamic voice jitter buffer (i.e. buffer delay period) is set based on the congestion information conveyed in the at least one of the plurality of VoIP packets for the call (col. 5, lines 13-23 and lines 39-47; col. 7, line 46 – col. 8, line 43; col. 11, lines 1-47).

Regarding claim 4, the packet switched communications system as in claim 1, wherein Schuster discloses at least one of the plurality of VoIP packets comprises:

at least one field that is set to indicate if the VoIP packet has traversed at least one router at a speed below a predetermined speed (col. 12, lines 24-55), and wherein the size of the dynamic voice jitter buffer is set based upon whether the at least one field is set (col. 7, line 62 – col. 8, line 44; col. 11, lines 1-47).

Regarding claim 5, the packet switched communications system as in claim 4, wherein Schuster discloses the at least one field in the VoIP packet is set with a congestion value (i.e. varying transmission delays; jitter) based upon a speed of an originating link (col. 9, line 66 - col. 10, line 28).

Regarding claim 6, the packet switch communications systems as in claim 4, wherein Schuster discloses:

the at least one field in the VoIP packet is set with a congestion value based upon at least one of a speed of a destination link, or a speed of a link immediately preceding the destination link (col. 12, lines 24-55).

Regarding claim 7, the packet switched communications system as in claim 1, wherein Schuster discloses the size of the dynamic voice jitter buffer (i.e. buffer delay period) is set to a first size if the congestion information is at or below a first threshold, and wherein the size of the dynamic voice jitter buffer is set to a second size if the congestion information is at or below a second threshold and above the first threshold, and wherein the size of the dynamic voice jitter buffer is set to a third size if the congestion information is above the second threshold (col. 10, lines 2-28; col. 11, lines 1-47).

Regarding claim 14, Schuster discloses a method for setting a size of a jitter buffer (i.e. buffer delay period) (Fig. 2, 34; i.e. jitter buffer; col. 3, lines 59-67)
for a call (col. 8, lines 44-48) for use with a packet switched communications system (Fig. 2; col. 1, lines 22-28; col. 2, lines 10-14) comprising the steps of:
receiving a plurality of voice over Internet protocol (VoIP) packets (i.e. real-time media; packet stream; col. 2, lines 10-14; col. 8, lines 43-56) for the call (col. 8, lines 48-52; col. 9, line 66 – col. 10, line 6), wherein at least one of the plurality of VoIP packets comprises a field for indicating an amount of transmission delay through a transmission path that the VoIP packet has encountered in the packet switched communications system, such that the field is set when at least one of the following events occurs: transmission rate for a link used for the VoIP packet is below a predetermined threshold or congestion of a link (i.e. varying transmission delays; jitter) exceeds a predetermined threshold (i.e. tolerable delay) (col. 10, lines 2-28);
determining whether the field of the VoIP packet is set (col. 10, lines 2-11); and
setting the size of the jitter buffer based upon whether the field is set in order to mitigate the effect of receipt of non-periodic VoIP packets at the destination device (col. 11, lines 1-47).

Regarding claim 16, the method as in claim 14, wherein Schuster discloses the field is set using a numeric value based upon the amount of transmission delay (col. 10, lines 2-11; col. 11, lines 1-40); and wherein the step of setting the jitter buffer comprises mapping the numeric value into a minimal jitter buffer size required for that amount of transmission delay (col. 11, lines 1-47).

Regarding claim 17, the method as in claim 16, wherein Schuster discloses the step of setting the size of the jitter buffer further comprises setting the jitter buffer (i.e. buffer delay period) to a first size if the numeric value is at or below a first value, and setting the size of the jitter buffer to a second size if the numeric value is at or below a second value and above the first value, and setting the size of the jitter buffer to a third size of the numeric value is above the second value (col. 10, lines 2-28; col. 11, lines 1-47).

Regarding claim 22, (see Schuster; col. 10, lines 12-28; Fig. 2, 34)

Regarding claim 23, Schuster discloses in a packet switched communications system (Fig. 2; col. 1, lines 22-28; col. 2, lines 10-14) having at least a source device (Fig. 2, 12; i.e. transmitter; col. 8, lines 44-48; col. 9, lines 7-49), a destination device (Fig. 2, 20; i.e. receiver; col. 8, lines 48-52; col. 9, line 66 – col. 10, line 6), and at least one router (i.e. processing device; router; col. 14, lines 33-51), the destination device comprising:

a receiver for receiving a plurality of voice over Internet protocol (VoIP) packets for a call from the source device via the at least one router (col. 8, lines 48-52; col. 9, line 66 – col. 10, line 6; col. 12, lines 35-42; col. 14, lines 33-51), wherein at least one of the plurality of VoIP packets for the call conveys congestion information (i.e. varying transmission delays; jitter)

to the destination device regarding the packet switched communications system (col. 10, lines 2-11); and

a jitter buffer (Fig. 2, 34; i.e. jitter buffer; col. 3, lines 59-67) for mitigating the non-periodic receipt of VoIP packets, wherein a size of the jitter buffer (i.e. buffer delay period) is set based on the congestion information conveyed in the at least one of the plurality of VoIP packets for the call (col. 10, lines 2-28; col. 11, lines 1-47).

Regarding claim 25, the destination device as in claim 23, wherein Schuster discloses at least one of the plurality of VoIP packets comprises at least one field that is set to indicate if the VoIP packet has traversed at least one router at a speed below a predetermined speed (col. 12, lines 24-55), and wherein the size of the jitter buffer is set based upon whether the at least one field is set (col. 7, line 62 – col. 8, line 44; col. 11, lines 1-47).

Regarding claim 26, the destination device as in claim 25, wherein Schuster discloses the at least one field in the VoIP packet is set with a congestion value (i.e. varying transmission delays; jitter) based upon a speed of an originating link (col. 9, line 66 - col. 10, line 28).

Regarding claim 27, the destination device as in claim 26, wherein Schuster discloses the size of the jitter buffer is set based upon the congestion value set within the at least one field (col. 7, line 62 – col. 8, line 44; col. 11, lines 1-47).

Regarding claim 28, the destination device as in claim 25, wherein Schuster discloses the at least one field in the VoIP packet is set with a congestion value based upon at least one of a speed of a destination link, or a speed of a link immediately preceding the destination link (col. 12, lines 24-55).

Regarding claim 29, the destination device as in claim 28, wherein Schuster discloses the size of the jitter buffer is based upon the congestion value set within the at least one field (col. 7, line 62 – col. 8, line 44; col. 11, lines 1-47).

Regarding claim 30, the destination device as in claim 23, wherein Schuster discloses the size of the jitter buffer (i.e. buffer delay period) is set to a first size if the congestion information is at or below a first threshold, and wherein the size of the jitter buffer is set to a second size if the congestion information is at or below a second threshold and above the first threshold, and wherein the size of the jitter buffer is set to a third size if the congestion information is above the second threshold (col. 10, lines 2-28; col. 11, lines 1-47).

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over Schuster as applied to claim 1 above, and further in view of U.S. Pat. Appl. Publ. 2002/0141392 by Tezuka et al, hereinafter Tezuka.

Regarding claim 2, the packet switched communications system as in claim 1, wherein Schuster does not disclose a time-to-live (TTL) field.

Tezuka discloses a packet switched communications system (Fig. 2; section: 0004, 0046-0047) having a dynamic voice jitter buffer (Fig. 2, 13; section 0054) for use with voice over Internet protocol (VoIP) packets (section 0004) comprising: a source (Fig. 2, 18; i.e. receiver

VoIP gateway apparatus) transmitting a plurality of VoIP packets (i.e. test packets; RTCP packets) (section 0051-0053);

a destination (Fig. 2, 12; Fig. 9, 22A; i.e. sender VoIP gateway apparatus) for receiving the plurality of VoIP packets (section 0051-0053, 0078); and

at least one router (i.e. intermediate router) for routing the plurality of VoIP packets for the call from the source to the destination (section 0053),

wherein at least one of the plurality VoIP packets conveys congestion information (i.e. packet arrival time jitter) regarding the packet switched communications system (section 0052), and

wherein a size of the dynamic voice jitter buffer (i.e. jitter buffer amount) is set based on the congestion information conveyed in the at least one of the plurality of VoIP packets (section: 0051-0054).

Wherein Tezuka discloses the at least one of the plurality of VoIP packets comprises a time-to-live (TTL) field that is set to a predetermined value (i.e. IP-TTL maximum value), wherein the predetermined value of the TTL field is decremented by one count each time the VoIP packet traverses a router in the packet switched communications system (section: 0053, 0078-0089), and wherein the size of the dynamic voice jitter buffer is set based on calculating the number of routers the VoIP packet has traversed (i.e. hop count) based on a final TTL value (i.e. IP-TTL value) determined at the destination (section 0053-0054, 0085, 0091).

Again, Schuster discloses the claimed system except Schuster determines the size of the dynamic voice jitter buffer based on congestion information in a least one VoIP packet. However, the claimed feature of a VoIP packet comprising a TTL field and the size of the buffer based on the value of the TTL field is taught by Tezuka.

It would have been obvious to one of the ordinary skill in the art at the time the invention was made to modify the system of Schuster to include the size of the dynamic voice jitter buffer is set based on the final TTL value as taught by Tezuka. One of ordinary skill in the art would have been lead to make include such a modification of Schuster to include a TTL field in a VoIP packet, such as the IP-TTL field of Tezuka, to the destination of Schuster to adjust the buffer at the destination based on the final TTL value of the VoIP packets. The benefit of providing a TTL field to the destination of Schuster as taught by Tezuka is that the size of the dynamic voice jitter buffer is determined by the number of routers the VoIP packet has hopped as revealed by the TTL field.

6. Claims 10, 12, and 13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Schuster in view of Tezuka.

Regarding claim 10, Schuster discloses a method for setting a size of a jitter buffer (Fig. 2, 34; i.e. jitter buffer; col. 3, lines 59-67) for a call for use in a packet switched communications system (Fig. 2; col. 1, lines 22-28; col. 2, lines 10-14) comprising the steps of: receiving a plurality of voice over Internet protocol (VoIP) packets (i.e. real-time media; packet stream) for the call (col. 8, lines 44-48; col. 9, lines 7-49), wherein at least one of the plurality of VoIP packets traverses a router (i.e. processing device; router) in the packet switched communication system (col. 14, lines 33-51);

setting the size of the jitter buffer based upon a field (i.e. varying transmission delays; jitter) in order to mitigate the effect of receipt of non-periodic VoIP packets at the destination device (col. 11, lines 1-47).

Schuster does not disclose a time-to-live (TTL) field.

Tezuka discloses a method for setting a size of a jitter buffer (Fig. 2, 13; section 0054) for use in a packet switched communications system (Fig. 2; section: 0004, 0046-0047) comprising the steps of: receiving a plurality of voice over Internet protocol (VoIP) packets (i.e. test packets; RTCP packets) (section 0051-0053, 0078), wherein at least one of the plurality of VoIP packets comprises a time-to-live (TTL) field in set to a predetermined value (i.e. IP-TTL maximum value) that is decremented by at least one count each time the VoIP packet traverses (i.e. hop count) a router (i.e. intermediate router) in the packet switched communication system (section 0053); reading a final value of the TTL field (i.e. IP-TTL value); and
setting the size of the jitter buffer based upon the final value of the TTL field in order to mitigate the effect of receipt of non-periodic VoIP packets at the destination device (section 0053-0054, 0085, 0091).

Again, Schuster discloses the claimed method except Schuster determines the size of the voice jitter buffer based on a field in a least one VoIP packet. However, the claimed feature of a VoIP packet comprising a TTL field and the size of the buffer based on the value of the TTL field is taught by Tezuka.

It would have been obvious to one of the ordinary skill in the art at the time the invention was made to modify the method of Schuster to include the size of the jitter buffer is set based on the final TTL value as taught by Tezuka. One of ordinary skill in the art would have been lead to make include such a modification of Schuster to include a TTL field in a VoIP packet, such as the IP-TTL field of Tezuka, to the destination device of Schuster to adjust the buffer at the destination device based on the final TTL value of the VoIP packets. The benefit of providing a TTL field to the destination device of Schuster as taught by Tezuka is that the size of the

dynamic voice jitter buffer is determined by the number of routers the VoIP packet has hopped as revealed by the TTL field.

Regarding claim 12, the method as in claim 10, wherein Schuster in view of Tezuka discloses the step of setting the size of the jitter buffer comprises:
comparing the predetermined value of the TTL field with the final value of the TTL field to produce a compared value; and setting the size of the jitter buffer based upon the compared value (Tezuka: section 0085, 0091).

Regarding claim 13, the method as in claim 12, wherein Schuster in view of Tezuka discloses the step of: setting the size of the jitter buffer (i.e. buffer delay period) further comprises setting the jitter buffer to a first size if the compared value is at or below a first value, and setting the size of the jitter buffer to a second size if the compared value is at or below a second value and above the first value, and setting the size of the jitter buffer to a third size if the compared value is above the second value (Schuster: col. 10, lines 2-28; col. 11, lines 1-47).

7. Claim 24 is rejected under 35 U.S.C. 103(a) as being unpatentable over Schuster as applied to claim 23 above, and further in view of Tezuka.

Regarding claim 24, the destination device as in claim 23, wherein Schuster does not disclose a time-to-live (TTL) field.

Tezuka discloses in a packet switched communications system (Fig. 2; section: 0004, 0046-0047) having at least a source device (Fig. 2, 18; i.e. receiver VoIP gateway apparatus), a destination device (Fig. 2, 12; Fig. 9, 22A; i.e. sender VoIP gateway apparatus), and at least one router (i.e. intermediate router; section 0053), the destination device comprising:

a receiver for receiving a plurality of voice over Internet protocol (VoIP) packets from the source device via the at least one router (section 0051-0053, 0078), wherein at least one of the plurality of VoIP packets conveys congestion information (i.e. packet arrival time jitter) to the destination device regarding the packet switched communications system (section 0052); and

a jitter buffer for mitigating the non-periodic receipt of VoIP packets, wherein a size of the jitter buffer (i.e. jitter buffer amount) is set based on the congestion information conveyed in the at least one of the plurality of VoIP packets (section: 0051-0054).

Wherein Tezuka discloses the at least one of the plurality of VoIP packets comprises a time-to-live (TTL) field that is set to a value (i.e. IP-TTL maximum value), wherein the value of the TTL field is decremented by one count each time the VoIP packet traverses a router in the packet switched communications system (section: 0053, 0078-0089), and wherein the size of the jitter buffer is set based on calculating the number of routers the VoIP packet has traversed (i.e. hop count) based on a final TTL value (i.e. IP-TTL value) determined at the destination device (section 0053-0054, 0085, 0091).

Again, Schuster discloses the claimed destination device except Schuster determines the size of the jitter buffer based on congestion information in a least one VoIP packet. However, the claimed feature of a VoIP packet comprising a TTL field and the size of the buffer based on the value of the TTL field is taught by Tezuka.

It would have been obvious to one of the ordinary skill in the art at the time the invention was made to modify the destination device of Schuster to include the size of the jitter buffer is set based on the final TTL value as taught by Tezuka. One of ordinary skill in the art would have been lead to make include such a modification of Schuster to include a TTL field in a VoIP

packet, such as the IP-TTL field of Tezuka, to the destination device of Schuster to adjust the buffer at the destination device based on the final TTL value of the VoIP packets. The benefit of providing a TTL field to the destination device of Schuster as taught by Tezuka is that the size of the jitter buffer is determined by the number of routers the VoIP packet has hopped as revealed by the TTL field.

Conclusion

8. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

9. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. See PTO-892 Form.

10. Any response to this action should be mailed to:

Commissioner for Patents
P.O. Box 1450

Alexandria, VA 22313-1450

Or faxed to:

(571) 273-8300 (for formal communications intended for entry)

Or call:

(571) 272-2600 (for customer service assistance)

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Lisa Hashem whose telephone number is (571) 272-7542. The examiner can normally be reached on M-F 8:30-5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Fan Tsang can be reached on (571) 272-7547. Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (571) 272-2600.

11. Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

LH
lh
February 16, 2008

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